(19) Europäisches Patentamt
European Patent Office
Office européen des brevets

(11) EP 0 703 689 A1

(12) EUROPEAN PATENT APPLICATION

(43) Date of publication:
27.03.1996 Bulletin 1996/13

(51) Int. Cl.$^{6}$: H04L 27/34

(21) Application number: 95111571.6

(22) Date of filing: 22.07.1995

(84) Designated Contracting States:
CH DE ES FR GB IT LI NL SE

(30) Priority: 28.07.1994 IT MI941626

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# (54) Method and system for transmitting and receiving digital signals, in which the peak power of the transmitted filtered signal is reduced

(57) The present invention relates to a method for improving the efficiency of a digital link, e.g. a microwave one. The invention is characterized by the utilization of a transmit coding (X) and a receive decoding for reducing the peak power of the filtered signal present at the input of the transmitter power amplifier, said coding being compatible with other conventional coding techniques (XI) for the increasing of the minimum distance between the sequences of transmitted points.

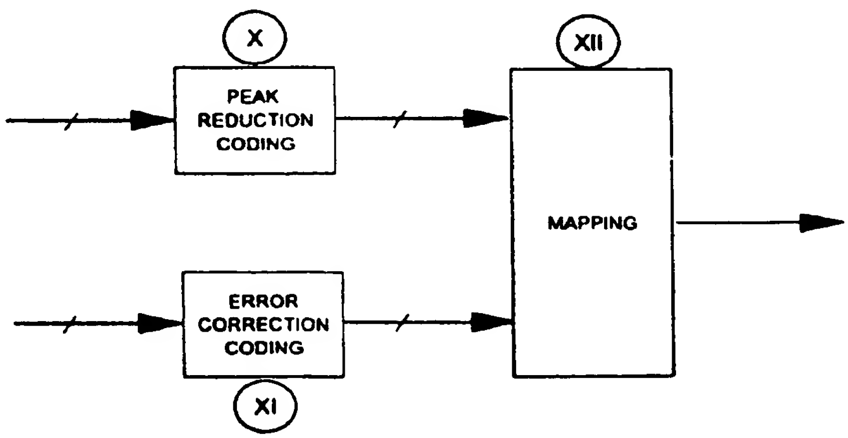


Figure 2

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## Description

Field of the invention

The present invention concerns a method and a system for the transmission and reception of numeric signals. In particular, it is used for improving the efficiency of a digital link, e.g. a microwave one.

Prior Art

For using the available band in an effective manner, the digital transmission systems make use of multilevel modulation techniques. The system threshold with respect to thermal noise increases with increasing the spectral efficiency and, in order to obtain a prefixed Bit Error Rate (hereinafter BER), greater values of the transmitted power are necessary. Such power is limited by the input-output characteristic of the transmitter final amplifier which is here structured as a nonlinear channel without storage.

When solid state amplifiers are used and/or when predistortion techniques without storage are utilized (see, e.g., G. Karam and H. Sari, "Analysis of predistortion, equalization and ISI cancellation techniques in digital radio systems with nonlinear transmit amplifiers", IEEE Transactions on Communications, vol. 37, No. 12, Dec. 1989), the input-output characteristic of the nonlinear channel (comprising in it also the effect of the amplifiers and of eventual predistortions) in practice is that of a hard limiter.

If a distortion-free transmission is desired, the peak of the complex envelope of the signal at the input of the nonlinear channel defined above must not go beyond the saturation point. The basic technique (see, e.g., T. Ryu, J. Uchibori and Y. Yoshida, "A stepped-square 256-QAM for digital radio systems", ICC '86, vol. 3, pages 1477-1481, June 1986) being used in order to avoid great values of the peak power, is to design the two-dimension constellation (hereinafter 2D) in such a way that the peak power of the unfiltered signal is as small as possible still observing the two restrictions of the assigned rate and of the minimum distance between the points of the transmitted constellation. An alternative approach (still observing the two restrictions mentioned above) is disclosed in the European Patent Application EP93111772, filed by the applicant, titled "Method and Apparatus for reducing the peak power of data sequences", inventors A. Sandri and A. Spalvieri, where it is emphasized that, to improve the efficiency of a digital link (e.g., a microwave one), it is useful to focus on the peak power of the filtered signal transmitted. In order to reduce such power, in the above mentioned patent application, A. Sandri and A. Spalvieri propose to use, in transmission, an encoder disposed between the modulator and the transmit filter, and, in reception, a decoder disposed between the receive adapted filter and the demodulator.

Summary of the invention

It is an object of the present invention to provide a method and a coding system that, on the same terms, reduce the peak power of the filtered signal present at the input of the transmitter power amplifier. It has been found, inter alia, that such a reduction is to advantage of radio relay links since it allows, for instance, the use of smaller antennas or the transmission over longer links. The object of the invention is achieved through a method as set forth in claim 1 and a circuit arrangement as set forth in claim 11. Further advantageous aspects of the invention are set forth in the dependent claims.

This invention provides an improvement over the above-mentioned patent application characterized by:

- a different and more efficient code construction, since it allows the peak value of the coded signal envelope to be further decreased while observing the above-mentioned constraints on the rate and on the minimum distance between the points of the transmitted constellation;
- a different arrangement of the encoder in transmission and of the decoder in reception, which allows the coding for the reduction of the filtered signal peak power to be effectively integrated with other conventional coding techniques for the increasing of the minimum distance between the sequences of transmitted points (e.g., of type Trellis Code Modulation, or of type Multi Level Code Modulation, see H. Imas and S. Hirakawa, "A new multilevel coding method using error correcting codes", IEEE Trans. on Inform. Th., vol. IT-23, pp 371-377, 1977 and G. Ungerboeck, "Trellis coded modulation with redundant signal sets", IEEE Communication Magazine, vol. 25, No. 2, Feb. 1987, for a description of such conventional coding techniques). In the absence of such compatibility it is possible to introduce only one of the two techniques mentioned above into the transmission system, and therefore realize only one of the two gains. In the case of the present invention, on the contrary, it is possible to introduce the two types of coding at the same time and obtain the sum of the gains relative to the individual codings as the overall gain.

The invention will now be illustrated in greater detail with reference to the attached drawings, in which:

- Fig. 1 shows the schematic block diagram of a generic digital transmission system, and

- Fig. 2 illustrates a possible embodiment of an encoder for the reduction of the peak power of a filtered signal.

Detailed description

Fig. 1 depicts the schematic block diagram of a generic digital transmission system; in this figure there are illustrated:

- a data source I (labelled DATA SOURCE) outputting the numeric sequence to be transmitted;
- a block II (labelled MODULATION & CODING) that receives at its input the numeric sequence to be transmitted and performs conventional coding operations to increase the minimum distance between the sequences of transmitted points (e.g., convolutional Trellis Code Modulation or Multi Level codings) and of modulation, outputting one of the points of the constellation to be transmitted. Inside this block an encoder is provided for reducing the peak power of the filtered signal in accordance with the invention;
- a transmission filter III ( labelled G(f) ) outputting the analog signal to be transmitted;
- a signal IV ( labelled x(t) ) at the input of the nonlinear portion of the transmitter;
- a block V (labelled NONLINEAR DISTORTION) representing an undesired non linear distortion on the signal path. It may be due to the non linear characteristic of the transmitter final amplifier (eventually accompanied with predistortion techniques) or, more in general, to a nonlinear behaviour of the transmission channel;
- a transmission channel proper VI (labelled LINEAR CHANNEL) outputting a signal formed by the signal at its input summed and/or combined with disturbances of various kind;
- a receive filter VII ( labelled "$G_{RX}(f)$" ) receiving the signal from the transmission channel and providing a suitable filtering;
- a block VIII (labelled DEMODULATION & DECODING) that receives at its input the filtered signal, demodulates it and carries out the decoding operations relative to the conventional codings mentioned above. Inside this block, a decoder for reducing the peak power in accordance with the invention is inserted;
- a user IX (labelled USER) receiving the numeric sequence.

In an advantageous and therefore preferred embodiment, the encoder for reducing the peak power of the filtered signal x(t) is designed and implemented as set forth hereinafter.

*Peak power of the encoded and filtered signal*

This section is devoted to evaluation of an overestimation (hereinafter: "upper bound") of the peak power of an encoded and filtered signal. This bound will be useful both for the construction of the code and for the evaluation of its features. The code to which we are interested has an infinite length. Let $M$ be the code and **m**=..., $m(-1)$, $m(0)$, $m(1)$, ...; $m(i) \in S$, be a codeword of $M$. The alphabet $S$ of the code is a constellation 2D, whose points ($s_1, s_2, \ldots, s_s$) are represented by complex numbers. The complex envelope at the output of the filter when m is presented at its input is

$$x(t) = m \otimes g(t) = \sum_{i=-\infty}^{\infty} m(i)\, g(t-iT) \qquad (1)$$

where g(t) is the impulse response of the transmission filter, T is the symbol interval and $\otimes$ is the convolution operator. The peak power of x(t) is

$$P_p(x) = \max_{-\infty \le t \le \infty} \{|x(t)|^2\} \qquad (2)$$

Assume that every version, shifted in time by integer multiples of T seconds, of a codeword is still a codeword under this assumption, the filtered signal x(t) is a cyclostationary process having period T. Since the period is T, eq. (2) is

equivalent to

$$P_p(M) = \max_{m\in M}\ \max_{\tau\le t<\tau+T}\{|m\otimes g(t)|^2\} \qquad (3)$$

Parameter τ is a time reference and can be arbitrarily fixed. Note that the evaluation of (3) is, in practice, not feasible until the code is truncated. Let n be the length of the truncated code, and C the truncated version of M. C is the union of all the n-tuples of labels of the codewords belonging to M. The codewords of C are denoted by **c** = c(1), c(2), ..., c(n), c(i) ∈ S. The following upper bound on the peak power of M applies

$$P_p(M) \le (\omega(S) + \max_{c\in C}\{\omega(c)\})^2 \qquad (4)$$

where

$$\omega(c) = \max_{\tau\le t<\tau+T}\{|c\otimes g(t)|\} \qquad (5)$$

and

$$\omega(S) = \max_{\tau\le t\le\tau+T}\{(\sum_{i=-\infty}^{0}|g(t-iT)| + \sum_{i=n+1}^{\infty}|g(t-iT)|)\}*\max_{s\in S}\{|s|\} \qquad (6)$$

The weight of ω(**c**) takes into account the effect of the codeword **c** on the instantaneous power of x(t) in the observation interval (τ, τ+T), while the weight ω(S) is an upper bound on the effect of the worse pair of queues. If a filter, physically realizable, is considered, the sums of the first term in eq. (6) converge to a finite value and therefore, in practice, they can be truncated. Since the upper bound (4) depends on τ, the leading to the minimum upper bound could be pursued. This procedure is long and of very poor help in reaching the main goal which is the code construction. The following τ' leading to an upper bound which is often very close to the minimum upper bound

$$\tau' \leftarrow \max_{-\infty\le\tau\le\infty}\{\int_{\tau-nT}^{\tau}|g(t)|^2dt\} \qquad (7)$$

The τ' indicated above is such that the maximum possible of energy of the pulse g(t) is in the interval (τ'-nT,τ'). When the waveform at the output of the filter is observed in the interval (τ', τ'+T), the truncated codeword **c** covers the interval (τ'-nT,τ'). If the interval (τ'-nT,τ') contains most of the energy of g(t), then the weight (6) is negligible with respect to weight (5), and therefore the choice τ=τ' leads to a value of (4) very close to the true peak power (3). To be noted expressly that equations (5) and (7) have been chosen according to an advantageous embodiment of the invention; they are susceptible of all those modifications and variations which, being apparent to those skilled in the art, are not described here. Lastly, equation (5) with τ=τ' represents the "weight" of the codeword **c**. This weight is computed as the highest peak in the complex envelope of the filtered signal x(t) that codeword **c** can generate when its relative time position with

respect to the impulse response of the filter g(t) is varied. The weight $\omega(\mathbf{c})$ is then interpreted in the following manner: the greater such weight, the worse the effect of codeword c on the peak power of the filtered signal x(t) will be.

Construction of the code

Let b/T be the constraint on the rate expressed in bits/second, i.e., let b denote the information bits to be transmitted every symbol interval T. Let S be the number of points of the constellation and let eq. $\log_2 S > b$, i.e. the number of points of the constellation is greater than the one strictly necessary to the trasmission of the b bits for each symbol interval T. The idea underlying the code construction is very simple and can be summarized as follows. Consider the universe code of infinite length, i.e. the set of all sequences of points from S. In the first step, all the sequences containing the n-tuples of higher weight $\omega(\mathbf{c})$ are discarded. In the second step, the sequences containing the n-tuples of higher weight are discarded from the code obtained in the first step, and so on. In this manner, the upper bound (4) on the peak power is decreased at each step. The discarding process can be continued until the code satisfies the rate constraint. In the limit, when the interval ($\tau'$-nT, $\tau'$) contains all the energy of g(t), the upper bound (4) equalizes the peak power and the discarding process ends with the code having the minimum peak power for the given initial constellation. The check whether the code, at the k-th discarding step, satisfies or less the rate constraint, is feasible through known graph theory techniques (see, e.g., B.H. Marcus, P.H. Siegel and J.K. Wolf, "Finite-state modulation codes for data storage", IEEE ISAC, vol. 10, No. 1 pp 5-37, Jan. 1992). Such techniques have been developed for realizing devices for recording data on magnetic support and, in the present invention, they are applied for the first time, to the multilevel numeric modulations. Hence, the construction of the code is formulated again in terms of graphs. The universe code is here represented by a graph A(0), consisting of $S^{n-1}$ states and S transitions that start from and converge in each state. Each state corresponds to a (n-1)-tuple of points of the constellation. The state a = ( a(1), a(2), ..., a(n-1) ), a(i) $\in$ S, is reached by the set of states (s,a(1),..., a(n-2) ), and starting from **a** it is possible to reach the set of states (a(2), a(3), ..., s). The weight $\omega(\mathbf{c})$ is the label of the transition starting from state ( c(1), c(2), ..., c(n-1) ) and ending in the state ( c(2), c(3), ..., c(n) ). To discard the sequences containing the n-tuple of higher weight, the transition of higher weight is to be simply discarded from the graph. The constraint on the rate is introduced by computing the capacity of the graph as described in the above-mentioned article by B.H. Marcus et al. Let $A^{(K)}$ be the graph after the k-th step in the construction of the code. The discard process stops at k=K such that $\mathrm{Cap}(A^{(K)}) \geq b$ and $\mathrm{Cap}(A^{(K+1)}) < b$. Note expressly that the use of the strategy set forth above allows the realization of codes more efficient than those proposed in the above-mentioned patent application, inventors A. Sandri and A. Spalvieri, since it allows - other conditions being equal - to discard a greater number of sequences of symbols, thus obtaining a further reduction of the peak power of the filter signal x(t) present at the input of the nonlinear portion of the transmitter. Once the graph describing the code has been defined as set forth above, it is possible to realize, by following mechanically known techniques disclosed in the above-mentioned article by B.H. Marcus et al., the encoder and the decoder to be inserted respectively in the transmitter and in the receiver. Such a procedure, in view of this article and of the foregoing description, is apparent to those skilled in the art and therefore it will be omitted for conciseness.

Sub-optimal codes

One way for reducing the design time and the complexity of the resulting encoder/decoder is to reduce the number of states of the initial graph. To this end, points from S can be grouped according to homogeneous groups. The term "homogeneous points" is used herein to indicate those points of the constellation having similar values of modulus and phase which therefore make a contribution about equal to the weight $\omega(\mathbf{c})$. Let G be the number of such groups. Now the universe code is described by a graph consisting of $G^{n-1}$ states and G groups of parallel transitions diverging from, and converging in, each state. Every group of parallel transitions is viewed as a unique entity whose label (weight) is the weight of the worse transition in the group (i.e. the higher weight among those associated with the transitions in the group). At each step of the code construction the group of transition having the worst weight is discarded. The idea underlying this discard-by-groups operation is that homogeneous groups of points of the constellation affect almost in the same way the peak power of the filtered signal, and therefore can be discarded or kept together. Compatibility with coded modulations Consider the case in which the constellation is partitioned into $2^{bc}$ subsets, and some code constraint is imposed on the sequences of subsets, while $b_u$ bits are left uncoded as often happens in conventional coded modulations devoted to the increasing of the minimum distance between the sequences of the transmitted points (see, e.g., the above-mentioned articles of H. Imai and S. Hirakawa and of G. Ungerboeck). In order for a peak reduction coding to be possible, the number of points in S must be greater than $2^{(bc + bu)}$. Compatibility of the peak reduction code and the conventional coded modulation in this circumstance means that the two code sequences are coded (and decoded) in an independent manner one from another. This means that the peak reduction coding must concern the uncoded bits b only. One way to do this is to group the points of the constellation in such a way that each group contains the same number of points from each subset. The construction of the code then goes on in the same way set forth in the sub-optimal codes section, the constraint on the rate being now (bc + bu)/T .

In an advantageous and, therefore, preferred embodiment, the peak reduction encoder is inserted inside block II of Fig. 1, in the manner shown in Fig. 2. In this figure there are:

- a block XI (labelled ERROR CORRECTION CODING) that receives at its input the bits relative to the choice of the subset and imposes on them a suitable conventional coding constraint (e.g., on the type Trellis Code Modulation or Multi Level Code Modulation) devoted to the increasing of the minimum distance between the sequences of the transmitted points;
- a block X (labelled PEAK REDUCTION CODING) along with the relative decoder in accordance with the invention that receives at its input the "uncoded" bits of the conventional coding and imposes on them a suitable coding constraint devoted to the peak power reduction, on the basis of what set forth in the previous sections (code construction and sub-optimal codes);
- a block XII (labelled MAPPING) providing at its output the symbol to be transmitted.

Some considerations about feature evaluation.

The features of a coded signal, transmitted on a channel with white gaussian additional noise under the peak power constraint, is expressed by the ratio

$$PDR = P_p(M)/\delta_{min}^{\ 2}$$

where $\delta_{min}^2$ is the minimum Euclidean square distance between any two points of the constellation at the decision element, and $g(t_0\text{-}t)$ is assumed to be the impulse response of the receive filter. Note that the proposed code construction does not optimize the PDR, since it does not take into account the term $\delta_{min}^2$. Therefore, we cannot guarantee an optimum in terms of PDR, not even for n---> ∞. As usual, the coding gain is the ratio between the PDR of a reference uncoded signal and the PDR reached by the coded one under the same rate constraint. This coding gain is interpreted as the increase of the signal-to-noise ratio at the decision element of a receiver with a filter adapted for a fixed value of the peak power at the input of the nonlinear channel defined above. Decoder The decoder (to be inserted in block VIII of Fig. 1, as said) can be defined starting from what has been set forth and defined in the foregoing sections, according to known rules, set forth in the above-mentioned article by B.H. Marcus et al. A detailed description thereof in the form of a block diagram is not given here (being apparent to those skilled in the art in the light of what described hereinbefore).

## Claims

1. Method for the transmission and the reception of numeric signals comprising the steps of:

   - modulating data from a numeric or made numeric source for the transmission in a transmit channel, by associating suitably data groups with points of a two-dimension or N-dimension constellation;
   - filtering the modulated signal;
   - transmitting the filtered signal through the transmit signal which is a nonlinear channel, where the nonlinearity may be due to the nonlinear characteristic of the final amplifier of the transmitter (eventually associated with predistortion techniques) and/or to a nonlinear behaviour of the real transmission channel; - receiving the outgoing signal from the transmit channel;
   - filtering the received signal;
   - demodulating the received signal for reconstructing the source data; characterized by:
   - coding in transmission so as to transmit most favourable sequences in terms of peak power of the filtered signal present at the input of the transmit channel.

2. Method according to claim 1, characterized in that the most favourable sequences are chosen in terms of the peak power of the filtered signal and such choice is carried out through a baseband numeric coding.

3. Method according to claim 1, characterized in that the coding is carried out in accordance with the following steps:

   - considering numeric sequences c having finite length;
   - calculating a weight w(c) relative to each sequence, said weight being computed as the highest peak in the complex envelope of the filtered signal that every sequence c is capable of generating when its relative time position is varied with respect to the impulse response of the transmission filter used;
   - arraying such sequences on the basis of said weight;
   - eliminating the sequences having greater weight;
   - repeating the elimination step an arbitrary number of times, provided that the resulting transmit capability of the system satisfies the constraint on the rate to be transmitted.

4. Method according to claim 3, characterized in that the weight w(c) is given by

$$\omega(c) = \max_{\tau \le t < \tau + T} \{ |c \otimes g(t)| \}$$

where

$$\tau' = \max_{-\infty \le \tau \le \infty} \{ \int_{\tau - nT}^{\tau} |g(t)|^2 dt \}$$

5. Method according to claim 3, characterized in that the coding further comprises the step of grouping the points of the constellation according to homogeneous groups (e.g. by modulus and phase) so as to reduce the number of states , the length of the code being equal.

6. Method according to the preceding claims, characterized in that said modulation is of type QAM (Quadrature Amplitude Modulation) or PSK (Phase Shift Keying).

7. Method according to the preceding claims, characterized by further comprising the steps of coding in transmission and decoding in reception said sequence through a conventional encoder for increasing the minimum distance between the sequences of the transmitted points other conditions being equal.

8. Method according to claim 7, characterized in that said steps of coding and decoding through a conventional encoder operate in a manner independent from the other steps.

9. Method according to claim 8, characterized in that the coding for reducing the peak power of the filtered signal x(t) operates on the "uncoded bits" of the conventional coding for increasing the minimum distance between the sequences of the transmitted points the other conditions being equal.

10. System for the transmit and receive of numeric signals from a numeric or made numeric source for transmission in a transmit channel, in a numeric sequence for implementing the method of the preceding claims, the system including :

- in transmission, a data source, an encoder, a modulator, a transmission filter and a nonlinearity;
- in reception, a filter, a decoder and a demodulator, characterized in that
- in transmission, an encoder for the transmission of the most favourable sequences inherent to the peak power of the filtered signal present at the input of the transmit channel, is inserted in the baseband portion of the system, between the data source and upstream of the transmission filter.

11. System according to claim 10, characterized in that it further comprises an encoder coding said sequences through a conventional encoder, i.e. designed - other conditions being equal - for the increasing of the minimum distance between the sequences of the transmitted points.

12. System according to claim 11, characterized in that said conventional encoder operates in a manner independent from the other steps.

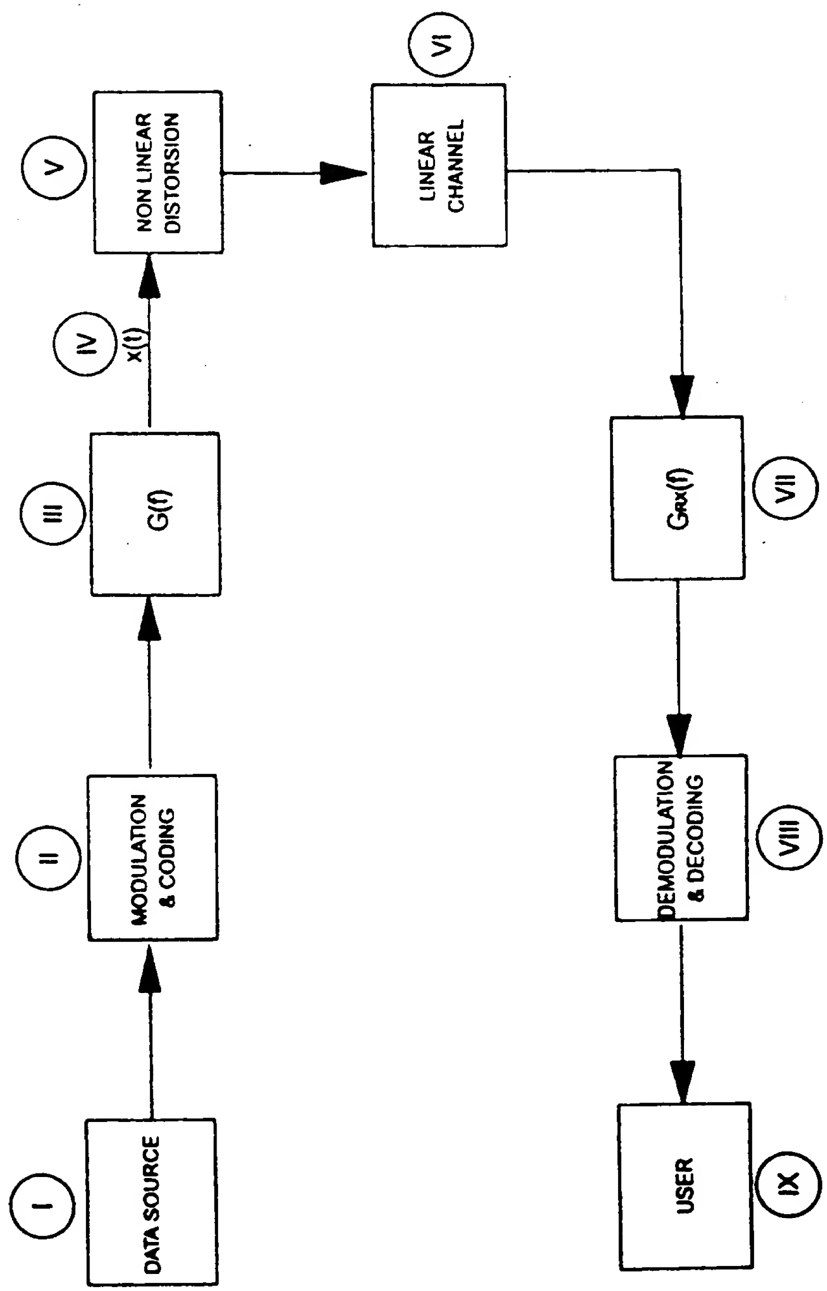
I
DATA SOURCE
II
MODULATION
& CODING
III
G(f)
IV
x(t)
V
NON LINEAR
DISTORSION
VI
LINEAR
CHANNEL
VII
$G_{RX}(f)$
VIII
DEMODULATION
& DECODING
IX
USER

Figure 1

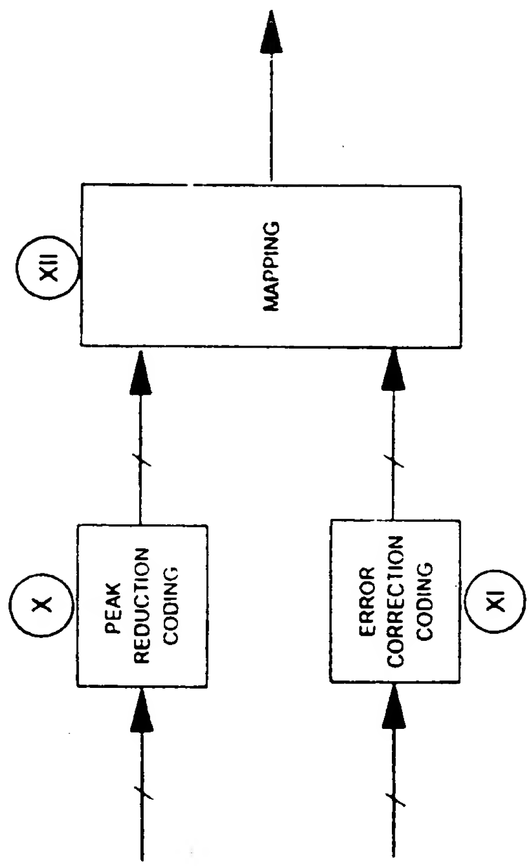
X
PEAK REDUCTION CODING
ERROR CORRECTION CODING
XI
XII
MAPPING

Figure 2

European Patent Office

# EUROPEAN SEARCH REPORT

Application Number
EP 95 11 1571

## DOCUMENTS CONSIDERED TO BE RELEVANT

| Category | Citation of document with indication, where appropriate, of relevant passages | Relevant to claim | CLASSIFICATION OF THE APPLICATION (Int.Cl.6) |
|---|---|---|---|
| X,D | EP-A-0 584 534 (ALCATEL ITALIA)<br>* page 3, line 27 - line 52 * | 1,2,6,7,10,11 | H04L27/34 |
| A | * figure 1 * | 3-5,8,9,12 | |
| A,D | IEEE JOURNAL ON SELECTED AREAS IN COMMUNICATIONS, JAN. 1992, USA, vol. 10, no. 1, ISSN 0733-8716, pages 5-37, MARCUS B. H. ET AL.: 'Finite-State Modulation Codes for Data Storage'<br>* the whole document * | 1-12 | |
| A | WO-A-92 17971 (BRITISH TELECOM)<br>* page 10, line 12 - page 20, line 28 * | 1-12 | |
| A | IEEE TRANSACTIONS ON INFORMATION THEORY, vol. 38, no. 2 PT.01, 1 March 1992 pages 281-300, XP 000257683 FORNEY JR G. D.: 'Trellis Shaping'<br>* section II, pages 283 - 287 * | 1-12 | TECHNICAL FIELDS SEARCHED (Int.Cl.6) |
| A | EUROPEAN TRANSACTIONS ON TELECOMMUNICATIONS, vol. 4, no. 3, 1 May 1993 pages 243-256, XP 000385751 VEDAT EYUBOGLU M. ET AL.: 'Advanced Modulation Techniques for V.Fast'<br>* paragraph 4.1, page 13 * | 1-12 | H04L |

The present search report has been drawn up for all claims

| Place of search | Date of completion of the search | Examiner |
|---|---|---|
| THE HAGUE | 7 November 1995 | Ghigliotti, L |

CATEGORY OF CITED DOCUMENTS

X : particularly relevant if taken alone
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